Allied Telesyn

Simply Connecting the World

AT-VP504E FXS (SIP/MGCP) VoIPTalk

User's Manual

PN 990-11591-10 Rev C

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Contents

Chapter 1

Using the AT-VP504E FXS 1

Before you Begin 1
Acronyms 2
Overview of the AT-VP504E FXS 2
Using the AT-VP504E FXS 3
Call Processes 3
Calls Involving Another Terminal 4
Calls Involving a Terminal and a LAN Endpoint 4
Calls Involving an Analog Gateway 5
Making Calls 7
Complete Dialing Sequence 7
Dialing a Telephone Number or a Numerical Alias 7
Dialing an IP Address 8
Using the Call Waiting Feature 9
Using the Call Transfer Feature 9
Call Transfer – Supervised 10
Call Transfer – Unsupervised 10
Using the Call Forward Feature 11
Call Forward Unconditional 11
Call Forward on Busy / No Answer 12
Conferencing Calls 13
Requirements 13
Managing a Conference Call 13

Appendix A

Glossary 15

1 Using the AT-VP504E FXS

Thank you for purchasing the AT-VP504E FXS from Allied Telesyn.

This manual illustrates some of the various call processes the AT-VP504E FXS supports. It also describes how to make and receive calls, as well as how to use the services the AT-VP504E FXS offers.

Before you Begin

This manual assumes that:

- your AT-VP504E FXS has been properly set up by your system administrator

  If you need to install and configure the AT-VP504E FXS yourself, please refer to the *Administration Manual* provided with your AT-VP504E FXS or contact your system administrator.

- the IP Communication Server has been properly installed and set up[1]

  If you need to install the IP Communication Server yourself, please refer to the *IP Communication Server Administration Manual* provided with your package.

Related Documentation

In addition to this Manual, each document set of the AT-VP504E FXS includes the following:

- IP Communication Server Manual[1]

  This manual explains how to install and configure the IP Communication Server, which is a set of software tools that helps manage a network of APA communication units and other SIP devices. It is intended for a network administrator. The manual is not printed – it is located on the Installation CD provided with your AT-VP504E FXS.

- *Administration Manual*

  This manual explains how to install and set up the various AT-VP504E FXS parameters. It is intended for a network administrator. The manual is not printed – it is located on the Installation CD provided with your AT-VP504E FXS.

1. Valid only if you have purchased the IP Communication Server.

- *AT-VP504E FXS Quick Start booklet*

  This printed booklet allows you to quickly setup and work with your AT-VP504E FXS.

- *IP Communication Server Quick Start booklet*[2]

  This printed booklet allows you to quickly setup and work with your IP Communication Server.

Acronyms

FXS	Foreign Exchange Service
IP	Internet Protocol
LAN	Local Area Network
LED	Light Emitting Diode
MAC	Media Access Control
Mb/s	Megabits per Second
MGCP	Media Gateway Control Protocol
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
SCN	Switched Circuit Network
SIP	Session Initiation Protocol
SNMP	Simple Network Management Protocol
TCP/IP	Transmission Control Protocol/Internet Protocol
TFTP	Trivial File Transfer Protocol
VoIP	Voice Over Internet Protocol
WAN	Wide Area Network

Overview of the AT-VP504E FXS

The AT-VP504E FXS is an IP Telephony adaptor that connects up to four (4) analog terminals to a LAN with access to an IP Network, allowing high-quality, full duplex, audio/fax communication.

This version of the AT-VP504E FXS can use either one of the following signalling protocol:

- the Session Initiation Protocol (SIP), which is a simple

2. Valid only if you have purchased the IP Communication Server.

signalling protocol for Internet conferencing and telephony.

- the Media Gateway Control Protocol (MGCP) version draft-huitema-megaco-mgcp-v0r1-05. MGCP is a protocol for controlling Voice over IP (VoIP) Gateways from intelligent external call control elements.

Using the AT-VP504E FXS

Now that your administrator has properly set up the AT-VP504E FXS and IP Communication Server configuration settings, you can dial any number on your phone (which is connected to the AT-VP504E FXS) and place the call.

However, you should be aware that the administrator has probably set permissions and restrictions regarding local and long distance calls. Should you encounter any calling problem, please discuss it with your administrator to remedy the problem and grant the necessary permissions.

Call Processes

The following examples illustrate some of the various calling processes the AT-VP504E FXS supports. These processes can be adapted at will to suit your needs and requirements.

The AT-VP504E FXS can communicate with the following devices:

- Another terminal on the IP network such as the AT-VP504E FXS.
- Any LAN Endpoint on the IP network such as:
  - a Soft Phone
  - an IP phone directly connected to the IP network
- A SCN phone or fax. However, the AT-VP504E FXS would need to contact an analog gateway such as the AT-VP504E FXO.

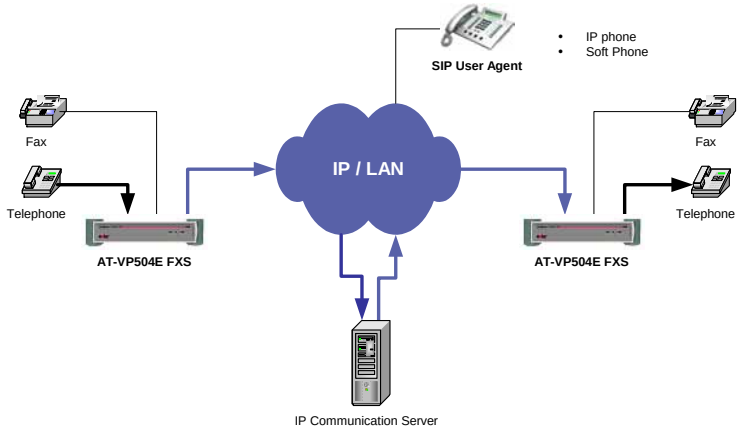
  For instance, the Switched Circuit Network (SCN) could be a Public Switched Telephone Network (PSTN) or some Private Branch eXchange (PBX).

Calls Involving Another Terminal

The following example illustrates how to reach a phone or fax on another AT-VP504E FXS terminal.

▶ Phone/Fax -> AT-VP504E FXS A -> AT-VP504E FXS B -> Phone/Fax

A user makes a call with the phone/fax connected to an AT-VP504E FXS, which in turn contacts another AT-VP504E FXS, then reaches the corresponding phone/fax.

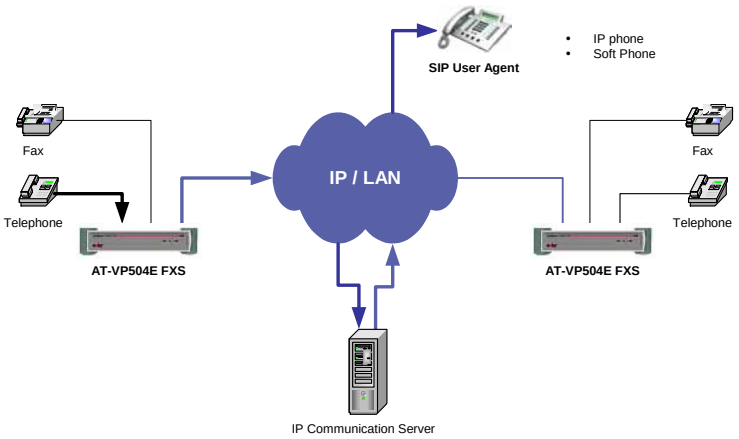


Calls Involving a Terminal and a LAN Endpoint

The following examples illustrate how a phone/fax connected to an AT-VP504E FXS terminal can communicate with a LAN Endpoint on the IP network.

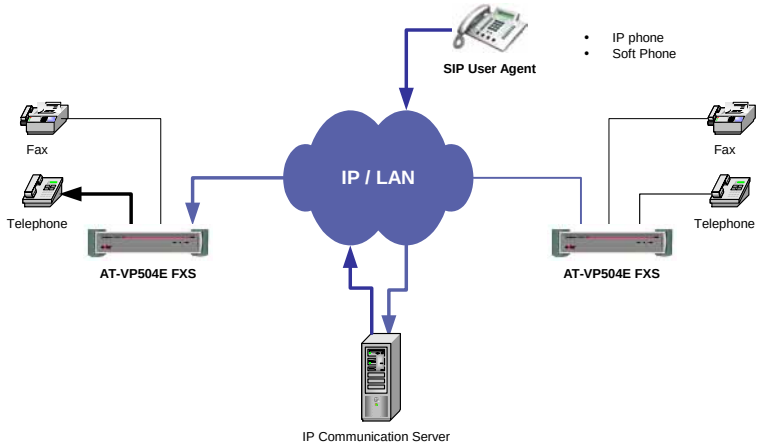
▶ Phone/Fax -> AT-VP504E FXS -> LAN Endpoint

A user makes a call with the phone/fax connected to an AT-VP504E FXS, which reaches the corresponding LAN Endpoint on the IP network.



▶ LAN Endpoint -> AT-VP504E FXS -> Phone/Fax

A LAN Endpoint contacts the AT-VP504E FXS, which reaches the corresponding phone/fax connected to the AT-VP504E FXS terminal.



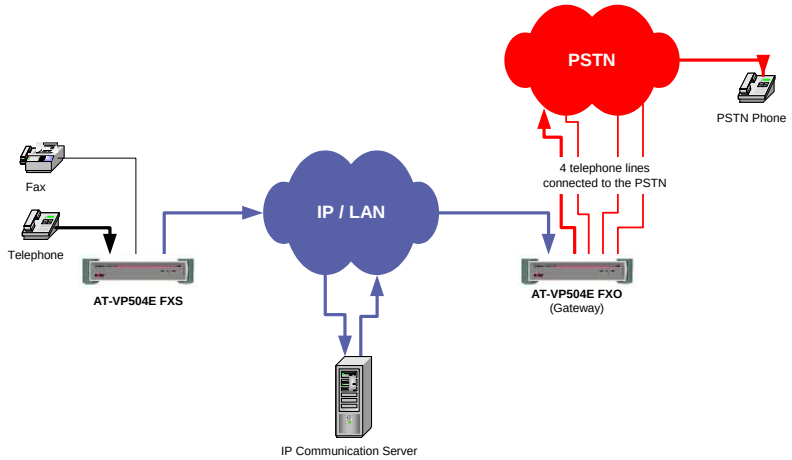
Calls Involving an Analog Gateway

The following example illustrates how a telephone/fax connected to an AT-VP504E FXS terminal and a SCN phone can communicate via an analog gateway.

▶ Phone/Fax -> AT-VP504E FXS -> AT-VP504E FXO (Gateway) -> SCN

A user makes a call with the phone/fax connected to an AT-VP504E FXS, which in turn contacts an AT-VP504E FXO gateway, then reaches the corresponding SCN Phone.

A SCN user can also contact the AT-VP504E FXO gateway, which in turn contacts the AT-VP504E FXS, then reaches the corresponding phone/fax.

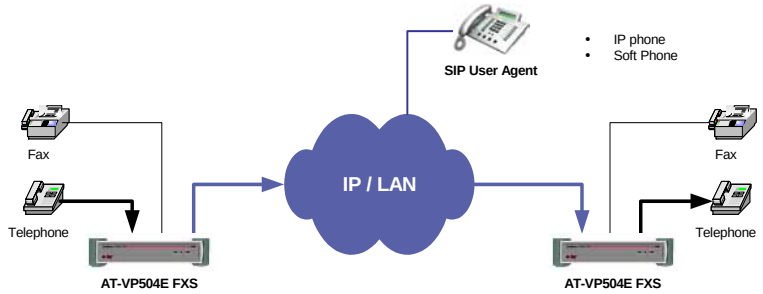


Calls Without a SIP Server

If the AT-VP504E FXS is not configured to register with a SIP Server[3], you can make SCN to IP network calls by dialing an IP address.

▶ IP Address Call

You can dial another communication unit (gateway or terminal) without the help of a SIP Server by entering its IP address. If you are dialing the IP address of an AT-VP504E FXS (which has four ports), by default, you will reach the telephone/fax connected to Port 1 of this 4 ports terminal.



Note: This type of dialing can only be possible if the AT-VP504E FXS is not configured to register with a SIP Server.

3. Only valid for units that run the SIP signalling protocol.

Making Calls

Users that have telephones or faxes connected to an AT-VP504E FXS will dial as if they were on a standard telephony system.

Complete Dialing Sequence

There are three ways to indicate the dialed number sequence is complete and the AT-VP504E FXS can dial the number:

- your administrator has set up the dialing process so that you must end your telephone number with a particular character to indicate it is complete, e.g. a "#".
- your administrator has set up the dialing process with a timer. This timer checks the dialing process and, when no further digits have been dialed for 4 seconds, it assumes the number is complete and dials it.
- your administrator has set up the AT-VP504E FXS so it knows exactly how many digits it must collect before it places the call. It finds the number of digits to collect by looking at the first few numbers dialed. For example: a telephone number beginning by 1 should be followed by 10 more digits in North America.

Consult your administrator to determine which dialing process is defined.

Dialing a Telephone Number or a Numerical Alias

This section assumes that the AT-VP504E FXS is configured to do SCN emulation. The AT-VP504E FXS could be configured to do any other kind of emulation, thus its users would simply have to dial as if they were using their old system.

To dial a Standard Call:

1. Dial the telephone number as if you were using a standard telephone, with country code and Area Code when required.

   Examples:

   ```
   829-8749
   1-514-570-1234
   ```

   A Standard Call uses the server to contact the remote dialed user. The server takes the decision as to redirect the call on the SCN or to keep it on the network. Keeping the call on the network takes precedence over redirecting it on the SCN. If

the call needs to go on the SCN, the server will redirect it to a proper analog gateway (such as the AT-VP504E FXO) that will place the call to the SCN network.

Note: You can dial one star numbers *xx (such as *69). These numbers will be automatically inserted in the Request-URL of the SIP INVITE request.

▶ **To dial a Forced SCN call:**

1. Dial "**".
2. Dial the telephone number as if you were using a standard telephone, with country code and Area Code when required.

   Examples:

   ```
   **829-8749
   **1-514-570-1234
   ```

   A Forced SCN Call allows you to specify that the user you want to reach is located on the SCN network. This leaves no decisions to the server; it must find a proper gateway and place the call on the SCN. This option can be useful only when a SCN number is shadowed by a network number. This type of call should not be used often.

Note: A forced SCN call will only be possible if an analog gateway such as the AT-VP504E FXO is available on the IP network.

Dialing an IP Address

You can dial another AT-VP504E FXS without the help of the IP Communication Server by entering its IP address. You will only be able to reach the phone or fax connected to Port 1 of this AT-VP504E FXS This procedure will only work if your AT-VP504E FXS is running the SIP signalling protocol. See your network administrator for more details.

▶ **To dial an IP address:**

1. Dial "***".
2. Dial the numerical digits of the IP address and use "*" for the "." of the IP address.
3. Dial "#" to terminate the IP address.

   For example, to dial 192.168.0.23, the user dials the following digits:

```
***192*168*0*23#
```

Using the Call Waiting Feature

The call waiting[4] feature allows you to put a call on hold. For instance, if you are already on the phone and a second call happens, a "beep" will be heard and repeated every ten (10) seconds to indicate there is a second incoming call.

The caller identity (the friendly name and phone number of the calling user) is displayed on telephones/faxes properly equipped with a LCD display.

To put the current call on hold:

1. Perform a Flash-Hook.

   This will put the call on hold and the second line is automatically connected to your line.

To switch from one line to the other:

1. Perform a Flash-Hook each time you want to switch between lines.

Note: It is important to answer a call on hold, or your second call will not have any answer.

To terminate the first call before answering the second call:

1. Hang up the phone.
2. Wait for the phone to ring.
3. Answer the phone.

   The second call is on the line.

Using the Call Transfer Feature

The call transfer[5] features allows you to transfer a current call to any other extension or phone number. There are two (2) types of call transfer features available:

- by consultation
- unsupervised

4. Only valid for units that run the SIP signalling protocol.
5. Only valid for units that run the SIP signalling protocol.

Call Transfer – Supervised

The call transfer – supervised allows you to transfer a current call to any other extension or phone number. However, the individual at the other extension or phone number must answer to complete the transfer.

▶ **To transfer a current call supervised:**

1. Perform a Flash-Hook.
   This will put the call on hold.
2. Wait for the transfer tone (three "beeps").
3. Dial the number to which you want to transfer the call.
   The third party will answer.
4. Hang up your phone.
   The call will be transferred.
5. If you want to get back to the first call (the call on hold), you must perform two Flash Hooks.
   You are back with the first call and the third party is released.

Note: If the number to which you want to transfer the call is busy or does not answer, quickly depress and release the plunger in or the actual handset-cradle twice. The busy tone or ring tone is cancelled and you are back with the first call.

Call Transfer – Unsupervised

The call transfer – unsupervised allows you to transfer a current call to any other extension or phone number. The individual at the other extension or phone number does not need to answer to complete the transfer.

▶ **To transfer a current call unsupervised:**

1. Perform a Flash-Hook.
   This will put the call on hold.
2. Wait for the transfer tone (three "beeps").
3. Dial the number to which you want to transfer the call.
4. Wait for the ringback tone, then hang up your phone.
   The call will be transferred. You can also wait for the third party to answer if you want. In this case, the call transfer becomes supervised.

5. If you want to get back to the first call (the call on hold), you must perform two Flash Hooks.

   You are back with the first call and the third party is released.

Using the Call Forward Feature

There are three types of Call Forward.

Call Forward Unconditional

The Call Forward – Unconditional[6] feature allows you to forward your calls to another extension or line.

When forwarding your calls outside the system, a brief ring will be heard on your phone to remind you that a call forward has been established. You can still make calls from your phone.

▶ **To forward calls:**

1. Take the receiver off-hook.
2. Wait for the dial tone.
3. Dial *25.

Note: The "*25" sequence can be user-configured. Check with your system administrator for the exact sequence actually implemented.

4. Wait for the transfer tone (three "beeps") followed by the dial tone.
5. Dial the number to which you want to forward your calls.

   Dial any access code if required.
6. Wait for three "beeps" followed by a silent pause.

   The call forward is established.
7. Hang up your phone.

   The calls are checked against the dial maps set up by your system administrator. See your system administrator for more information.

6. Only valid for units that run the SIP signalling protocol.

▶ **To check if the call forward has been properly established:**

1. Take the receiver off-hook.
2. Wait for the dial tone.
3. Dial your extension or phone number.

   The call is forwarded to the desired phone number.
4. Hang up your phone.

▶ **To cancel the call forward:**

1. Take the receiver off-hook.
2. Wait for the dial tone.
3. Dial *26.

Note: The "*26" sequence can be user-configured. Check with your system administrator for the exact sequence actually implemented.

4. Wait for the transfer tone (three "beeps") followed by the dial tone.

   The call forward is cancelled.
5. Hang up your phone.

Call Forward on Busy / No Answer

You can automatically forward incoming calls[7] to a pre-determined extension within your system if you do not answer before a specific number of rings or if you are already on the line.

Note: The call forward on busy / no answer can only be set up by your system administrator. See your system administrator for more information.

7. Only valid for units that run the SIP signalling protocol.

Conferencing Calls

A conference call[8] between three parties can be set up.

Requirements

For the conference call to occur succesfully, all parties must meet the following requirements:

- Support at least one of the PCM codecs (G.711 μ-law and G.711 A-law) enabled on the port that is having the conference.
- Ability to dynamically change codec during a call.

Managing a Conference Call

If you are on the phone with one person and want to conference with a third one, you can do so. In the following examples, let's assume that:

- A is the conference initiator.
- B is the person called on the first line.
- C is the person called on the second line.

▶ To initiate a three-way conference (A and B already connected):

1. A performs a Flash-Hook.

   This will put B on hold and the second line is automatically connected. A hears a dial tone.

2. A dials C's number.

   A and C are now connected.

3. A performs another Flash-Hook.

   The call on hold (B) is reactivated. A is now conferencing with B and C.

▶ A wants to transfer B to C during the conference:

1. A hangs up.

   The conference is terminated. B and C are now connected.

8. Only valid for units that run the SIP signalling protocol.

▶ **A wants to terminate the call with C and get back to the call with B during the conference:**

1. A performs a Flash-Hook.

   The conference is terminated and the call with C is disconnected. A and B are still connected and can go on with their conversation.

▶ **B (or C) hangs up during the conference:**

1. B (or C) hangs up during the conference.

   The conference is terminated, but the call between A and C (or B) is not affected and they are still connected.

Area Code The preliminary digits that a user must dial to be connected to a particular outgoing trunk group or line. In North America, an Area Code has three (3) digits and is used with a NXX (office code) number. For example, in the North American telephone number *561-955-1212*, the numbers are defined as follows:

Table 1: North American Numbering Plan

No.	Description
561	Area Code, corresponding to a geographical zone in a non-LNP (Local Number Portability) network.
955	NXX (office code), which corresponds to a specific area such as a city region.
1212	Unique number to reach a specific destination.

Outside North America, the Area Code may have any number of digits, depending on the national telecommunication regulation of the country. In France, for instance, the numbering terminology is defined as *xZABPQ 12 34*, where:

Table 2: France Numbering Plan

No.	Description
x	Operator forwarding the call. This prefix can be made of 4 digits.
Z	(regional) geographical zone of the number (in France, there are 5 zones). It has two (2) digits.
ABPQ	First 4 digits corresponding to a local zone defined by central offices.
12 34	Unique number to reach a specific destination.

In this context, the Area Code corresponds to the *Z* portion of the numbering plan. Since virtually every country has a different dialing plan nomenclature, it is recommended to identify the equivalent of an Area Code for the location of your device.

CC Acronym for *Country Code*.

1. In international direct telephone dialing, a code that consists of 1-, 2-, or 3-digit numbers in which the first digit designates the region and succeeding digits, if any, designate the country.
2. In international record carrier transmissions, a code consisting of 2- or 3-letter abbreviations of the country names, or 2- or 3-digit numbers that represent the country names, that follow the geographical place names.

Dual-Tone Multi-Frequency (DTMF) In telephone systems, multi-frequency signaling in which a standard set combinations of two specific voice band frequencies, one from a group of four low frequencies and the other from a group of four higher frequencies, are used. Although some military telephones have 16 keys, telephones using DTMF usually have 12 keys. Each key corresponds to a different pair of frequencies. Each pair of frequencies corresponds to one of the ten decimal digits, or to the symbol "#" or "*", the "*" being reserved for special purposes.

Dynamic Host Configuration Protocol (DHCP) TCP/IP protocol that enables PCs and workstations to get temporary or permanent IP addresses (out of a pool) from centrally-administered servers.

Flash-Hook Quickly depressing and releasing the plunger in or the actual handset-cradle to create a signal to a PBX or Centrex that special instructions will follow such as transferring the call to another extension.

FXS Line Foreign Exchange Service/Station. A network-provided service in which a telephone in a given local exchange area is connected, via a private line, to a central office in another, i.e., "foreign", exchange, rather than the local exchange area's central office. A FXS line is normally connected to a standard telephone, fax or modem.

Gateway A device that links two different types of networks that use different protocols (for example, between the packet network and the Public Switched Telephone Network).

IP Acronym for *Internet Protocol*. The IP protocol is a standard describing software that keeps track of the Internet's addresses for different nodes, routes outgoing messages, and recognises incoming messages.

Light Emitting Diode (LED)	A semiconductor diode that emits light when a current is passed through it.
Local Area Network (LAN)	Data-only communications network confined to a limited geographic area, with moderate to high data rates. Contrast with WAN.
Network	A group of two or more computer systems that are linked.
Off-hook	A line condition caused when a telephone handset is removed from its cradle.
On-hook	A line condition caused when a telephone handset is resting in its cradle.
Packet	Group of bits transmitted as a complete package on a packet-switched network.
Port	Network access point, the identifier used to distinguish among multiple simultaneous connections to a host.
Private Branch Exchange (PBX)	A small to medium sized telephone system and switch that provides communications between onsite telephones and exterior communications networks.
Protocol	Defines a common set of rules and signals that computers on the network use to communicate.
Public Switched Telephone Network (PSTN)	The local telephone company network that carries voice data over analog telephone lines.
Server	A computer or device on a network that works in conjunction with a client to perform some operation, for example a Windows NT Server.
Session Initiation Protocol (SIP)	SIP is a simple signaling protocol for Internet conferencing and telephony.
Single Network Monitor Protocol (SNMP)	SNMP is the protocol governing network management and the monitoring of network devices and their functions.

Switched Circuit Network (SCN)	A communication network, such as the public switched telephone network (PSTN), in which any user may be connected to any other user through the use of message, circuit, or packet switching and control devices.
T.38	An ITU-T Recommendation for Real-time fax over IP. T.38 addresses IP fax transmissions for IP-enabled fax devices and fax gateways, defining the translation of T.30 fax signals and Internet Fax Protocols (IFP) packets.
Telephony	The science of translating sound into electrical signals, transmitting them, and then converting them back into sound.
Terminal	Device capable of sending or receiving data over a data communications channel.
Transmission Control Protocol/Internet Protocol (TCP/IP)	A suite of communications protocols developed by the Department of Defense in the 1970s that connects hosts on the Internet and provides the standards for transmitting data over networks.
Trivial File Transfer Protocol (TFTP)	Allows for transferring files (such as software upgrades) from a remote device.
Voice Over IP (VoIP)	The technology used to transmit voice conversations over a data network using the Internet Protocol. Such data network may be the Internet or a corporate Intranet.
Wide Area Network (WAN)	A computer network where the computers are not geographically close and are linked by telephone lines or radio waves.

Index

A

acronyms 2

C

call
- a LAN Endpoint 4
- another terminal 4
- forced SCN 8
- IP address 6, 8
- standard 7
- without SIP Server 6

call forward
- on busy / no answer 12
- unconditional 11

call transfer
- supervised 10
- unsupervised 10

call waiting, using 9
conferencing 13

D

DHCP 16
dialing
- a telephone number 7
- forced SCN call 8
- IP address call 6, 8
- standard call 7

DTMF 16

F

Flash-Hook 16
FXS Line 16

G

Gateway 16

H

hold, putting a call on 9

I

IP address, dialing 6, 8

L

LAN 17
LED 17

M

making
- forced SCN call 8
- IP address call 6, 8
- standard call 7

O

Off-hook 17
On-hook 17

P

Packet 17
Port 17
Private Branch Exchange (PBX) 17
Protocol 17
PSTN 17

S

Server 17
SIP 17
SNMP 17

T

T.38 18
TCP/IP 18
telephone number, dialing 7
TFTP 18
transferring a call
- supervised 10
- unsupervised 10

V

VoIP 18